PATENT COOPERATION TREATY

From the INTERNATIONAL BUREAU

PCT

NOTIFICATION OF ELECTION

(PCT Rule 61.2)

To:

United States Patent and Trademark Office
(Box PCT)
Crystal Plaza 2
Washington, DC 20231
ÉTATS-UNIS D'AMÉRIQUE

in its capacity as elected Office

Date of mailing (day/month/year) 20 April 1999 (20.04.99)	
International application No. PCT/SG97/00037	**Applicant's or agent's file reference** SGS/48414
International filing date (day/month/year) 29 August 1997 (29.08.97)	**Priority date** (day/month/year)
Applicant SAPNA, George et al	

1. The designated Office is hereby notified of its election made:

   ☒ in the demand filed with the International Preliminary Examining Authority on:

   02 March 1999 (02.03.99)

   ☐ in a notice effecting later election filed with the International Bureau on:

2. The election ☒ was ☐ was not

   made before the expiration of 19 months from the priority date or, where Rule 32 applies, within the time limit under Rule 32.2(b).

The International Bureau of WIPO **34, chemin des Colombettes** **1211 Geneva 20, Switzerland** Facsimile No.: (41-22) 740.14.35	Authorized officer S. Mafla Telephone No.: (41-22) 338.83.38

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INTERNATIONAL PRELIMINARY EXAMINATION REPORT

(PCT Article 36 and Rule 70)

Applicant's or agent's file reference SGS/48414	FOR FURTHER ACTION	See Notification of Transmittal of International Preliminary Examination Report (Form PCT/IPEA/416)
International application No. PCT/SG97/00037	International filing date *(day/month/year)* 29/08/1997	Priority date *(day/month/year)* 29/08/1997
International Patent Classification (IPC) or national classification and IPC H04H1/00		
Applicant SGS-THOMSON MICROELECTRONICS ASIA ...... et al.		

1. This international preliminary examination report has been prepared by this International Preliminary Examining Authority and is transmitted to the applicant according to Article 36.

2. This REPORT consists of a total of 6 sheets, including this cover sheet.

   ☒ This report is also accompanied by ANNEXES, i.e. sheets of the description, claims and/or drawings which have been amended and are the basis for this report and/or sheets containing rectifications made before this Authority (see Rule 70.16 and Section 607 of the Administrative Instructions under the PCT).

   These annexes consist of a total of 15 sheets.

3. This report contains indications relating to the following items:

   I ☒ Basis of the report
   II ☐ Priority
   III ☐ Non-establishment of opinion with regard to novelty, inventive step and industrial applicability
   IV ☐ Lack of unity of invention
   V ☒ Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations suporting such statement
   VI ☐ Certain documents cited
   VII ☒ Certain defects in the international application
   VIII ☐ Certain observations on the international application

Date of submission of the demand	Date of completion of this report
02/03/1999	07. 12. 99
Name and mailing address of the international preliminary examining authority: European Patent Office D-80298 Munich Tel. +49 89 2399 - 0 Tx: 523656 epmu d Fax: +49 89 2399 - 4465	Authorized officer von der Straten, G Telephone No. +49 89 2399 8994

INTERNATIONAL PRELIMINARY EXAMINATION REPORT

International application No. PCT/SG97/00037

I. Basis of the report

1. This report has been drawn on the basis of (*substitute sheets which have been furnished to the receiving Office in response to an invitation under Article 14 are referred to in this report as "originally filed" and are not annexed to the report since they do not contain amendments.*):

Description, pages:

13	as originally filed			
1-12	as received on	26/07/1999	with letter of	20/07/1999

Claims, No.:

1-13	as received on	26/07/1999	with letter of	20/07/1999

Drawings, sheets:

1/4-4/4	as originally filed

2. The amendments have resulted in the cancellation of:

- ☒ the description, pages: 13
- ☐ the claims, Nos.:
- ☐ the drawings, sheets:

3. ☐ This report has been established as if (some of) the amendments had not been made, since they have been considered to go beyond the disclosure as filed (Rule 70.2(c)):

4. Additional observations, if necessary:

V. Reasoned statement under Article 35(2) with regard to novelty, inventive step or industrial applicability; citations and explanations supporting such statement

1. Statement

Novelty (N)	Yes:	Claims	1-13
	No:	Claims	
Inventive step (IS)	Yes:	Claims	2-10
	No:	Claims	1, 11-13
Industrial applicability (IA)	Yes:	Claims	1-13
	No:	Claims	

2. Citations and explanations

see separate sheet

VII. Certain defects in the international application

The following defects in the form or contents of the international application have been noted:

see separate sheet

1. Reference is made to the following documents:

   D1 = EP, A, 0 564 089
   D2 = EP, A, 0 506 111
   D3 = EP, A, 0 590 790

3. **Concerning item V**

a. Document D1, see in particular the abstract and pages 2,3,6 and 17, discloses, according to essential features of **claim 1**, a method of decoding digital audio data (see page 1, lines 9 - 11: efficient encoding and decoding of audio signals; see page 6, lines 41 - 46: PCM digital audio signals) which already comprises the step of obtaining an input sequence of data elements representing encoded audio samples (page 6, lines 51 - 53: frames), preprocessing the input sequence of data elements (page 17, lines 32 - 39 and figure 12: box labelled "Read & Decode") performing a modified discrete cosine transform (see the abstract) and forming decoded audio signals (page 17, lines 32 - 39 and figure 12: Stereophonic Decoder, box labelled "Inverse quantization and reconstruction of right and left channels").

The subject-matter of claim 1 differs from that disclosed in D1 in the way in which the processing of input data is performed.

Since the person skilled in the art of coding and decoding of real time data streams has always the aim of reducing the amount of processing required for the coding and, in particular, for the decoding process, the person skilled in the art would search the relevant state of the art to find a transform which requires only a reduced number of arithmetic operations, and, hence would find document D2.

Document D2, which is directed to a data processing method for video data discloses, see in particular the abstract and page 8, lines 15 - 37 and figure 5, the method steps of calculating an array of sum data (page 8, lines 27 - 30: z_k = (x0+x7), (x1+x6), (x2+x5), (x3+x4)) and an array of difference data (page 8, line 31: w_k = (x0-x7), (x1-x6), (x2-x5), (x3-x4)), calculating a first sequence of output values using the array of sum data (page 8, lines 32 - 37: parallel multiplication circuits 6a to 6d carry out an operation in accordance with data z_k), calculating a

second sequence of output values using the array of difference data (page 8, lines 32 - 37: parallel multiplication circuits 6e to 6h carry out an operation in accordance with data w_k).

It thus would be obvious to the person skilled in the art, namely when the same result is to be achieved, ie. to reduce the amount of processing required for decoding, to apply these features with corresponding effect to a method according to document D1, thereby arriving at a method according to claim 1.

The subject-matter of **claim 1** does therefore not involve an inventive step (Article 33(3) PCT).

b. The arrangement of **claim 11** corresponds to the method of claim 1 and once the principle of the method of claim 1 is available to the skilled person as demonstrated above with regard to D1 and D2, the structural details defined by **claim 11** for implementing the method of claim 1 are also considered as falling within the design capability of a skilled person and cannot offer a basis for an inventive claim.

c. Dependent **claims 12 and 13** do not appear to contain any additional features which, in combination with the features of any claim to which they refer, involve an inventive step for the following reasons:

The essential feature of **claim 12,** the use of the inverse modified discrete cosine transform, is disclosed in document D3, see page 2, lines 3 - 17: IMDCT.

The essential feature of **claim 13,** the use of the means disclosed in claims 11 and 12 in an MPEG decoder, is disclosed in document D1, see page 3, lines 11 - 15: MPEG-Audio Psychoacoustic II Model.

Therefore, the subject-matter of **claims 12 and 13** does not involve the required inventive step, Article 33 (3) PCT.

4. **Concerning item VII**

The independent claims should have been drafted in the proper two-part "characterised" form recommended by Rule 6.3.(b),(i),(ii) PCT, having a preamble that correctly reflects the nearest prior art, presumably that represented by the above noted **D1**.

In order to meet the requirements of Rule 5.1.(a),(ii) PCT, the relevant prior art, i.e. the documents D1 and D2 noted above, should have been acknowledged by reference and briefly discussed in the introductory part of the description.

The claims do not include reference signs in parentheses where features shown in the drawings are referred to, Rule 6.2.(b) PCT.

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PATENT COOPERATION TREATY

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INTERNATIONAL SEARCH REPORT

(PCT Article 18 and Rules 43 and 44)

Applicant's or agent's file reference SGS/48414	FOR FURTHER ACTION see Notification of Transmittal of International Search Report (Form PCT/ISA/220) as well as, where applicable, item 5 below.	
International application No. PCT/SG 97/ 00037	International filing date *(day/month/year)* 29/08/1997	(Earliest) Priority Date *(day/month/year)*
Applicant SGS-THOMSON MICROELECTRONICS ASIA ...... et al.		

This International Search Report has been prepared by this International Searching Authority and is transmitted to the applicant according to Article 18. A copy is being transmitted to the International Bureau.

This International Search Report consists of a total of 3 sheets.

[X] It is also accompanied by a copy of each prior art document cited in this report.

1. [ ] **Certain claims were found unsearchable** (see Box I).

2. [ ] **Unity of invention is lacking** (see Box II).

3. [ ] The international application contains disclosure of a **nucleotide and/or amino acid sequence listing** and the international search was carried out on the basis of the sequence listing
   - [ ] filed with the international application.
   - [ ] furnished by the applicant separately from the international application,
     - [ ] but not accompanied by a statement to the effect that it did not include matter going beyond the disclosure in the international application as filed.
   - [ ] Transcribed by this Authority

4. With regard to the **title**,
   - [ ] the text is approved as submitted by the applicant
   - [X] the text has been established by this Authority to read as follows:

FAST SYNTHESIS SUB-BAND FILTERING METHOD FOR DIGITAL SIGNAL DECODING

5. With regard to the **abstract**,
   - [X] the text is approved as submitted by the applicant
   - [ ] the text has been established, according to Rule 38.2(b), by this Authority as it appears in Box III. The applicant may, within one month from the date of mailing of this International Search Report, submit comments to this Authority.

6. The figure of the **drawings** to be published with the abstract is:

   Figure No. 2
   - [ ] as suggested by the applicant.
   - [X] because the applicant failed to suggest a figure.
   - [ ] because this figure better characterizes the invention.
   - [ ] None of the figures.

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International Bureau

INTERNATIONAL APPLICATION PUBLISHED UNDER THE PATENT COOPERATION TREATY (PCT)

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(71) Applicant (*for all designated States except US*): SGS-THOMSON MICROELECTRONICS ASIA PACIFIC (PTE) LTD. [SG/SG]; 28 Ang Mo Kio Industrial Park 2, Singapore 569508 (SG).

(72) Inventors; and
(75) Inventors/Applicants (*for US only*): SAPNA, George [IN/SG]; Block 315, 2 Serangoon Avenue #06-220, Singapore 550315 (SG). YANG, Haiyun [CN/SG]; Block 501, 5 Ang Mo Kio Avenue #10-3702, Singapore 560501 (SG).

(74) Agent: DONALDSON & BURKINSHAW; P.O. Box 3667, Singapore 905667 (SG).

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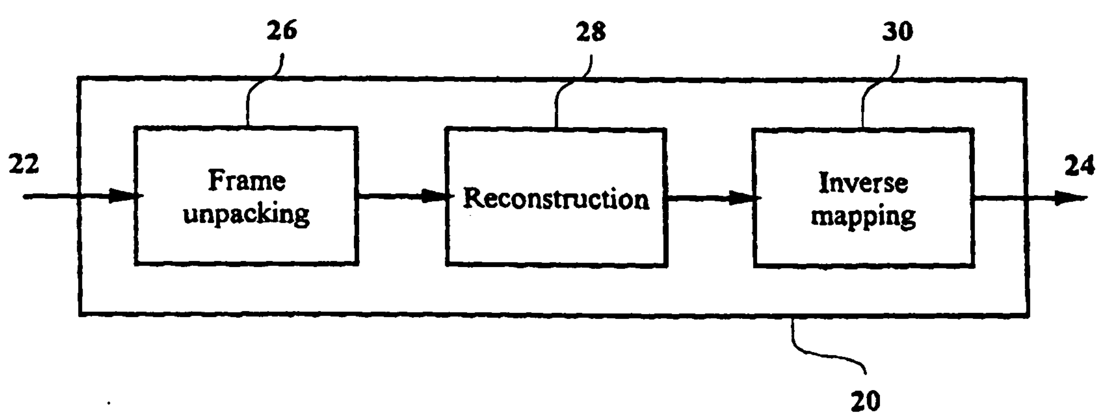
(54) Title: FAST SYNTHESIS SUB-BAND FILTERING METHOD FOR DIGITAL SIGNAL DECODING



(57) Abstract

In order to reproduce audio signals which have been compressed or encoded for storage or transmission using, for example, MPEG audio encoding, a synthesis sub-band filter is employed which performs an inverse modified discrete cosine transform (IMDCT). The computational cost of the IMDCT implementation is reduced by pre-calculating arrays of sum and difference data. The arrays of sum and difference data are then used in two separate transform calculations, the results of which can be used in the generation of pulse code modulation (PCM) audio data.

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FAST SYNTHESIS SUB-BAND FILTERING METHOD FOR DIGITAL SIGNAL DECODING

This invention relates to digital signal decoding for the purposes primarily of audio reproduction. In particular, the invention relates to enhanced synthesis sub-band filtering during decoding of digital audio signals.

In order to store or transmit data representing audio signals it is often desirable to first encode or compress the data so as to enable it to be stored or transmitted more efficiently. Decoding the data requires that the stored or transmitted data be reconstructed into audio signals by application of a decoding or decompression technique. The reconstruction process is typically quite computationally intensive, yet the process should be fast and reliable enough to enable the audio signals to be reconstructed in real time, on the fly, for example. In order for the decoding process to be carried out in relatively low-cost consumer products, the hardware utilised by the decoder should also preferably be relatively simple and inexpensive, or at least to the greatest extent reasonably possible.

Efficient stereo and multichannel digital audio signal coding methods have been developed for storage or transmission applications such as Digital Audio Broadcasting (DAB), Integrated Service Digital Network (ISDN), High Definition Television (HDTV) and Set Top Box (STB) for video-on-demand. The formats used to encode and reciprocally decode digital audio and video information for storage and retrieval is subject to various standards, one of which has been established by the Moving Pictures Experts Group and is known as the MPEG standard. A standard on low bit rate coding for mono or stereo audio signals was established by MPEG-1 Audio, published under ISO-IEC/JTC1 SC29 11172-3, entitled "Coding of Moving Pictures and Associated Audio for Digital Storage Media at up to About 1.5 Mbit/s", and the disclosure of that document is incorporated herein by reference. MPEG-2 Audio (ISO/IEC 13818-3) provides the extension to 3/2 multichannel audio and an optional low frequency enhancement channel (LFE). The audio part of the standard, ISO/IEC 11172-3, defines three algorithms, Layer 1, 2 and 3 for coding PCM audio signals. MPEG-2 (Multichannel) also defines Layer 1, 2, and 3 algorithms.

The MPEG audio encoder processes a digital audio signal and produces a compressed bitstream for transmission or storage. The encoder algorithm is not standardised, and may use various means for encoding such as estimation of the auditory masking threshold, quantisation, and scaling. However, the encoder output must be such that a decoder conforming to the above-mentioned standards specification will produce audio suitable for the intended application.

The decoder, subject to the application-dependent parameters, accepts the compressed audio bitstream in the defined syntax, decodes the data elements and uses the information to produce digital audio output, also according to the defined standard. The decoder first unpacks the received bitstream to recover the encoded audio information frame by frame. After the process of frame unpacking, the decoder performs an inverse quantisation (expansion process) and feeds a sub-band synthesis filter bank with a set of 32 scaled-up sub-band samples in order to reconstruct the output PCM audio signals. The sub-band filter banks used for Layer 1 and Layer 2 of MPEG 1 audio decoder and Layer 1 and Layer 2 of MPEG2 (Multichannel extension) audio decoder, are the same.

The sub-band synthesis filter is one of the most computationally intensive blocks of the MPEG audio decoder. Sub-band filtering is performed for each sub-band in a frame and for every channel. Any reduction in its computational requirements thus enables less complexity and reduced cost of decoding.

In accordance with the present invention there is provided a method of decoding digital audio data, comprising the steps of obtaining an input sequence of data elements representing encoded audio samples, calculating an array of sum data and an array of difference data using selected data elements from the input sequence, calculating a first sequence of output values using the array of sum data, calculating a second sequence of output values using the array of difference data, and forming decoded audio signals from the first and second sequences of output data.

Preferably, the array of sum data is obtained by adding together respective first and second data elements from the input sequence, the first and second data elements being selected from mutually exclusive sub-sequences of the input sequence. Furthermore, the array of difference data is preferably obtained by subtracting respective first data elements from corresponding second data elements of the input sequence, the first and second data elements being selected from mutually exclusive sub-sequences of the input sequence.

In one form of the invention the step of calculating an array of sum data and an array of difference data comprises dividing the input data sequence into first and second equal sized sub-sequences, the first sub-sequence comprising the high order data elements of the input sequence and the second sub-sequence comprising the low order data elements of the input sequence, calculating the array of sum data by adding together each respective data element of the first sub-sequence with a respective corresponding data element of the second sub-sequence, and calculating the array of difference data by subtracting each respective data element of the first sub-sequence from a respective corresponding data element of the second sub-sequence.

The invention also provides method of decoding a sequence of m, m an even positive integer, input digital audio data samples S[k], where k = 0, 1, ... (m-1), to produce a set of n, n an even positive integer, output audio data samples V[i], where i = 0, 1, ...(n-1), comprising the steps of:

a) calculating an array of sum data $S_{ADD}[k]$ according to

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

b) calculating an array of difference data $S_{SUB}[k]$ according to

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

c) calculating a first output audio data sample by a multiply-accumulate operation according to

$$V[2i] = V[2i] + N[2i, k]*S_{ADD}[k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

$$\text{where } N[2i, k] = \cos\left[\frac{(32+2i)(2k+1)\pi}{64}\right]$$

d) calculating a second output audio data sample by a multiply-accumulate operation

according to

$$V[2i+1] = V[2i+1] + N[2i+1, k]*S_{SUB}[k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

$$\text{where } N[2i+1, k] = \cos\left[\frac{(32+(2i-1))(2k+1)\pi}{64}\right]$$

e) and repeating steps c) and d) for i = 0, 1, ... (n/2-1) to obtain a full set of output data.

The invention further provides a synthesis sub-band filter for use in decoding digital audio data, comprising a means for receiving or retrieving an input sequence of data elements comprising encoded digital audio data, a pre-calculation means for calculating an array of sum data and an array of difference data using selected data elements from the input sequence, and a transform calculation means for calculating a first sequence of decoded output values using said array of sum data and a second sequence of decoded output values using said array of difference data.

The invention is described in greater detail hereinbelow, by way of example only, with reference to the accompanying drawings, in which:

Figure 1 is a block diagram of major functional portions of an MPEG audio encoder;

Figure 2 is a block diagram of major functional portions of an MPEG audio decoder;

Figure 3 is a flow diagram of an MPEG decoding procedure;

Figure 4 is a flow diagram showing a generalised form of a procedure according to the present invention; and

Figure 5 is a flow diagram illustrating a preferred implementation of the invention.

Figure 1 is a block diagram illustrating the major components of an MPEG audio encoder circuit 2 constructed in accordance with the aforementioned standards document. In the figure, an input signal 4, comprising a pulse code modulated (PCM) signal having a 48 kHz sampling frequency and a sample size of 16 bits per sample, is provided as input to the single channel encoder 2. The input signal is first mapped from the time domain into the frequency domain by a sub-band filter bank 8. The resulting coefficients are normalized with scale factors which may be transmitted as side information. The coefficients thus obtained are then

quantized and entropy encoded by a quantizer and encoding circuit 10. Masking thresholds of the quantization errors are calculated based on psychoacoustic values provided by a psychoacoustic model 14 to control the quantization step. The bit allocation is transmitted as side information. The coded signal is then multiplexed by a frame packing circuit 12 and an encoded bitstream 6 is produced at the output of the encoder 2.

A block diagram illustrating the main components of an MPEG audio decoder circuit 20 is shown in Figure 2. In the figure, an encoded bitstream 22 is provided to the input of the decoder. A bitstream unpacking and decoding circuit 26 performs an error correction operation if such operation was applied in the encoder. The bitstream data are unpacked to recover the various pieces of encoded information, and a reconstruction circuit 28 reconstructs the quantized version of the set of mapped samples from the frames of input data. An inverse mapping circuit 30 transforms the mapped samples back into a uniform pulse code modulated (PCM) output signal 24 that reproduces the corresponding input signal which was provided to the encoder.

The foregoing descriptions of the encoder and decoder are specific to the MPEG standard, and it is considered to be within the skill of those in the art to implement the various hardware functions described above. Accordingly, a more detailed hardware description of an MPEG coding system is not considered necessary for a full and complete understanding of the invention. It should be appreciated the invention described herein, although described in connection with the MPEG coding standard, is considered useful for other coding applications and standards.

Referring to Figure 3, there is shown a flow diagram 40 of steps involved in signal processing in layers I and II in an MPEG1 audio decoder. To begin with, the bit allocation of an input bitstream (42, 44) is decoded (46). Thereafter, various scale factors are also decoded (48) and the samples are requantized (50). The encoded signal is decoded in a synthesis sub-band filter (52) and the decoded pulse code modulated signals are output (54, 56) for further processing and/or real time reproduction. The present invention relates

primarily to the synthesis sub-band filter portion of the decoding process, when implemented for MPEG decoding.

The synthesis sub-band filter bank is composed of two main functions, an Inverse Modified Discrete Cosine Transform (IMDCT) and an Inverse Pseudo-Quadrature Mirror Filter (IPQMF). The IMDCT, which can be viewed as an overlap transform, performs a 32 x 64 cosine modulation transformation, which means a frequency shift of a filter bank into one single filter.

Consider a system in which output sub-band audio signal samples V_i (i=0....63) are decoded from sequences of 32 encoded input samples S_k, k = 0....31. The inverse MDCT of the sequence S_k, is defined as follows:

$$V_i=\sum_{k=0}^{31}\cos\left[\frac{(16+i)(2k+1)\pi}{64}\right]*S_k \quad for\ i=0,1,.....63 \tag{1}$$

Taking the cosine symmetric property wherein:

$$\cos\theta=\cos(2\pi-\theta) \tag{2}$$

the IMDCT definition equation (1) may be modified as given below to implement a 32-point IMDCT. The remaining 32 output audio signal samples are obtained after post-processing from this IMDCT of S.

$$V_i=\sum_{k=0}^{15}\cos\left[\frac{(32+i)(2k+1)\pi}{64}\right]*\left[S_k+(-1)^i*S_{31-k}\right] \quad for\ i=0,1,.....31 \tag{3}$$

This equation (3) may be computed according to the following algorithm:

repeat i = 32 times

repeat k = 16 times

if I is even, Sum = S[k] + S[31-k]

if I is odd, Sum = S[k]- S[31-k]

*V[i] = V[i] + N[i, k] * Sum*

end k

end i

where i		*is the index of output samples ($i=0....31$)*
	k	*is the index of input samples ($k=0....15$)*
	$N(i,k) =$	$\cos\left[\frac{(32+i)(2k+1)\pi}{64}\right]$
	$S[k]$	*represents the input sample data sequence*
	$V[i]$	*represents the output of IMDCT*

The IMDCT equation, making use of the symmetrical property, is given in Equation (3) above, and the computational effort required for MPEG audio decoding is in large part dependant upon the efficiency with which the input samples can be processed through the IMDCT to obtain respective sub-band filter PCM samples. Embodiments of the present invention are able to reduce the number of arithmetic operations performed in implementing the IMDCT portion of the decoder, to thereby increase the computational efficiency of the decoding process. In particular, the number of addition operations required for the implementation of this equation can be reduced substantially by pre-computing the sum and difference of the sample data which is the input to the IMDCT. In addition, the pre-computation can take place outside the main IMDCT computational loop. Hence the main loop contains only the MAC operations, which can be executed very efficiently by any general purpose DSP in a minimum number of cycles.

In the present invention, the dequantised sample data (e.g. 32 samples) from the encoded bitstream is pre-processed as per the symmetrical property of the cosine coefficients. The sample data is then split into two banks, each containing 16 samples. The sum and difference of respective data elements in the two banks is computed and stored in two arrays. These

arrays are used as the input data for the subsequent MAC operations.

Prior art implementations of equation (3) have required 32 x 16 Multiply-Accumulate operations and 32 x 16 Addition operations. By using the pre-computation operations described above, however, the number of Addition operations reduces to 2 x 16. This results in a saving of 30 x 16 Addition operations per Sub-band filter implementation, which in turn translates to a corresponding reduction in overall computational power.

In the IMDCT equation (3), S_k represents a sequence of m input data samples, where k = 0 ... (m-1). In a typical implementation for MPEG decoding 32 input data samples may be processed, such that m=32. For pre-computing the sum and difference of respective data elements, the input data sample sequence is first arranged into two equally sized data banks, one constituting the high order data elements and the other the low order data elements:

Data bank (1)	S_k	for k = 0 ... (m/2)-1
Data bank (2)	Sk	for k = (m/2) ... (m-1)

For example, in a preferred embodiment of the present invention where m=32, S_k is split into two data banks comprising:

(1)	S_k	for k = 0 .. 15
(2)	S_k	for k = 16 .. 31

The sum and difference data are calculated using respective data elements from the two data banks and is stored in two arrays of data, S_{ADD} and S_{SUB}, which are computed as follows:

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad for\ k = 0,\ 1,\ ....(m/2)-1 \qquad (4)$$

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad for\ k = 0,\ 1,\ ....(m/2)-1 \qquad (5)$$

In the aforementioned example of 32 input data samples, equations (4) and (5) reduce to:

$S_{ADD}[k] = S[k] + S[31-k]$ for k = 0, 1, ...15

$S_{SUB}[k] = S[k] - S[31-k]$ for k = 0, 1, ...15

The IMDCT equation (3) may now be divided into two portions and rewritten as follows:

$$V[i] = \sum_{k=0}^{15} \cos\frac{(32+i)(2k+1)\pi}{64} * S_{ADD}[k] \quad \text{for } i = 0,2,4,\ldots.30 \tag{6}$$

$$V[i] = \sum_{k=0}^{15} \cos\frac{(32+i)(2k+1)\pi}{64} * S_{SUB}[k] \quad \text{for } i = 1,3,5,\ldots..31 \tag{7}$$

As shown in the above equations (6) and (7), the IMDCT may now be calculated in two passes, an 'even pass' where the sum of the sample data is used (equation (6)), and an 'odd pass' where the difference of the sample data is used (equation (7)). The computational algorithms of the above equations are shown below.

Calculation of sum and difference of sample data (Addition operations)

repeat k = 16 times

$S_{ADD}[k] = S_k + S_{31-k}$

$S_{SUB}[k] = S_k - S_{31-k}$

end k

Calculation of 'even' data of IMDCT (Multiply-Accumulate operations)

repeat i = 16 times

repeat k = 16 times

$V[i] = V[i] + N[i,k] * S_{ADD}[k]$

end k

end i

Calculation of 'odd' data of IMDCT (Multiply-Accumulate operations)

```
repeat i = 16 times
    repeat k = 16 times
        V[i] = V[i] + N[i,k]*S_SUB[k]
    end k
end i
```

where	i	*is the index of output samples (i=0...31)*
	k	*is the index of input samples (k=0....15)*
	$N(i,k) =$	$\mathrm{Cos}\left[\frac{(32+i)(2k+1)\pi}{64}\right]$
	$S[k]$	*represents the input sample data sequence*
	S_{ADD}	*represents the sum of data array*
	S_{SUB}	*represents the difference of data array*
	$V[i]$	*represents the output of the IMDCT*

Figures 4 and 5 illustrate the above procedure according to a preferred embodiment of the invention in the form of flow diagrams. The representation shown in Figure 4, illustrates the general steps involved, and the procedure illustrated in the flow diagram 80 of Figure 4 corresponds to the synthesis sub-band filter step 52 of the overall decoding procedure 40 of Figure 3. To begin with the input samples S_k are received (82, 84) after having been isolated from the frames of encoded data received or retrieved. The input data samples are then utilised for pre-calculation of sum and difference data, as described above. This involves dividing the input data sample set into two equal sized sub-sets, which in the preferred embodiment consists of a first sub-set comprising the lower order data and a second sub-set comprising the higher order data. For example, in the case of 32 input samples S_0 to S_{31} as described, the first sub-set of input sample data may comprise the lower order input data S_0 to S_{15} and the second sub-set comprises the upper order data samples S_{16} to S_{31}. Respective ones of each sub-set of input sample data are then used to obtain a sets of sum and difference data, S_{ADD} and S_{UB}. As can be readily ascertained from the above description, in the preferred embodiment the calculation of the sum and difference data is performed using the

lowest order samples from the first set with the corresponding highest samples from the second set. For example, in the case of 32 input samples, the sum and difference data elements may be calculated as follows:

$$S_{ADD}[0] = S[0] + S[31] \qquad S_{SUB}[0] = S[0] - S[31]$$
$$S_{ADD}[1] = S[1] + S[30] \qquad S_{SUB}[1] = S[1] - S[30]$$
$$S_{ADD}[2] = S[2] + S[29] \qquad S_{SUB}[2] = S[2] - S[29]$$
$$\vdots \qquad \vdots$$
$$S_{ADD}[15] = S[15] + S[16] \qquad S_{SUB}[15] = S[15] - S[16]$$

Once the arrays of sum and difference data have been calculated, the multiply-accumulate operations required to calculate the IMDCT can be performed iteratively in two steps. The first step (88) is used to obtain half of the output samples (e.g. the "even" outputs) using the pre-calculated sum data comprising the S_{ADD} data elements. The second step (90) is used to obtain the other half of the output samples (e.g. the "odd" outputs) using the pre-calculated difference data comprising the S_{SUB} data elements. Each of these steps (88, 90) is an iterative multiply-accumulate (MAC) operation involving each of the data elements from the respective S_{ADD} or S_{SUB} array. Furthermore, each of the MAC operations of steps 88, 90 are performed repeatedly (step 92) to obtain a full complement of output samples. For example, where 32 output samples V_0 to V_{31} are required, each of the iterative MAC steps 88, 90 would be performed 16 times. Once the data for each output has been calculated, the data samples are output for PCM processing (step 94).

A more detailed preferred embodiment of the decoding procedure is illustrated in the flow diagram 100 shown in Figure 5. Beginning at step 102, a sequence of m input samples S_k (k = 0 .... m-1) are received for decoding to n sub-band filter outputs V_i (i = 0 .... n-1) at step 104. In the preferred embodiment for an MPEG implementation, both the number of input samples m and the number of output samples n are the same, 32. Steps 106, 108 and 110 of procedure 100 form a loop for the pre-calculation process of determining and storing the sum

and difference data arrays from the input data samples. The steps 112, 114, and 116 then form nested loops for the iterative multiple-accumulate calculation of the "even" ones of the output data elements (e.g. V_i for $i = 0, 2, 4, \ldots 30$), using the pre-calculated sum data array S_{ADD}. A calculation loop of steps 112 and 114 provides the iterative MAC operation, whilst the loop provided by step 116, enables calculation of each (even) alternate output data element. The remaining (odd) alternate output data elements are calculated in nested loop steps 118, 120, 122 using the difference data array S_{SUB}. The resulting output sub-band data is then provided at final step 124.

The preferred form of the invention presented herein results in a reduction of 480 addition operations per 32 sub-band samples. For a stereo output MPEG1 Layer 2 audio decoder, this is a reduction of 480 *36*2 arithmetic operations per frame. The overall reduction in arithmetic operations which is achieved is approximately 46.875% per IMDCT.

It will be readily apparent to those of ordinary skill in the relevant art that the present invention may be implemented in numerous different ways, without departing from the spirit and scope of the invention as described herein, and it is to be understood that such modifications are considered to be within the scope of the invention. In any event, it is immediately recognisable that one way the invention can be carried out, relating as it does to the processing of data, is using general purpose computing apparatus operating under the instruction of software or the like which is produced separately and specially adapted to perform the methods of the invention. Alternatively, specialised computing apparatus such as a dedicated integrated circuit, chipset or the like may be constructed with the functions of the invention embedded therein. Many other variations to the particular implementation will of course be possible. It will also be recognised that in places in the description and appended claims where it is said that a data set is divided into sub-sets, for example, this division may be simply a notional one, and no physical separation need occur, as is known in the data processing art.

The foregoing detailed description of the present invention has been presented by way of

example only, and is not intended to be considered limiting to the invention which is defined in the claims appended hereto.

CLAIMS:

1. A method of decoding digital audio data, comprising the steps of obtaining an input sequence of data elements representing encoded audio samples, calculating an array of sum data and an array of difference data using selected data elements from the input sequence, calculating a first sequence of output values using the array of sum data, calculating a second sequence of output values using the array of difference data, and forming decoded audio signals from the first and second sequences of output data.

2. A method as claimed in claim 1, wherein the array of sum data is obtained by adding together respective first and second data elements from the input sequence, the first and second data elements being selected from mutually exclusive sub-sequences of the input sequence.

3. A method as claimed in claim 1 or 2, wherein the array of difference data is obtained by subtracting respective first data elements from corresponding second data elements of the input sequence, the first and second data elements being selected from mutually exclusive sub-sequences of the input sequence.

4. A method as claimed in claim 1, wherein the step of calculating an array of sum data and an array of difference data comprises dividing the input data sequence into first and second equal sized sub-sequences, the first sub-sequence comprising the high order data elements of the input sequence and the second sub-sequence comprising the low order data elements of the input sequence, calculating the array of sum data by adding together each respective data element of the first sub-sequence with a respective corresponding data element of the second sub-sequence, and calculating the array of difference data by subtracting each respective data element of the first sub-sequence from a respective corresponding data element of the second sub-sequence.

5. A method as claimed in claim 1, wherein the step of calculating a first sequence of

output values comprises performing a multiply-accumulate operation utilising each of the sum data elements.

6. A method as claimed in claim 1 or 5, wherein the step of calculating a second sequence of output values comprises performing a multiply-accumulate operation utilising each of the difference data elements.

7. A method as claimed in any preceding claim wherein the input sequence of data elements is derived from MPEG encoded audio data, and wherein the decoded audio signals comprise pulse code modulation samples.

8. A method of decoding a sequence of m, m an even positive integer, input digital audio data samples S[k], where k = 0, 1, ... (m-1), to produce a set of n, n an even positive integer, output audio data samples V[i], where i = 0, 1, ...(n-1), comprising the steps of:

a) calculating an array of sum data $S_{ADD}[k]$ according to

$$S_{ADD}[k] = S[k] + S[m-1-k] \qquad \text{for } k = 0, 1, \ldots(m/2-1)$$

b) calculating an array of difference data $S_{SUB}[k]$ according to

$$S_{SUB}[k] = S[k] - S[m-1-k] \qquad \text{for } k = 0, 1, \ldots(m/2-1)$$

c) calculating a first output audio data sample by a multiply-accumulate operation according to

$$V[2i] = V[2i] + N[i, k]*S_{ADD}[k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

$$\text{where } N[i, k] = \cos\left[\frac{(32+2i)(2k+1)\pi}{64}\right]$$

d) calculating a second output audio data sample by a multiply-accumulate operation according to

$$V[2i+1] = V[2i+1] + N[i, k]*S_{SUB}[k] \qquad \text{for } k = 0, 1, \ldots (m/2-1)$$

$$\text{where } N[i, k] = \cos\left[\frac{(32+(2i+1))(2k+1)\pi}{64}\right]$$

e) and repeating steps c) and d) for i = 0, 1, ... (n/2-1) to obtain a full set of output data.

9. A method as claimed in claim 8, wherein the number m of input digital audio data

samples is 32, and the number n of output audio data samples is 32.

10. A method as claimed in claim 8 or 9, wherein the decoding steps are repeated for decoding a series of frames of encoded audio data in an MPEG format.

11. A synthesis sub-band filter for use in decoding digital audio data, comprising a means for receiving or retrieving an input sequence of data elements comprising encoded digital audio data, a pre-calculation means for calculating an array of sum data and an array of difference data using selected data elements from the input sequence, and a transform calculation means for calculating a first sequence of decoded output values using said array of sum data and a second sequence of decoded output values using said array of difference data.

12. A synthesis sub-band filter as claimed in claim 11 wherein the pre-calculation means and transform calculation means collectively perform an inverse modified discrete cosine transform of the encoded digital audio data.

13. An MPEG decoder including a synthesis sub-band filter as claimed in claim 11 or 12.

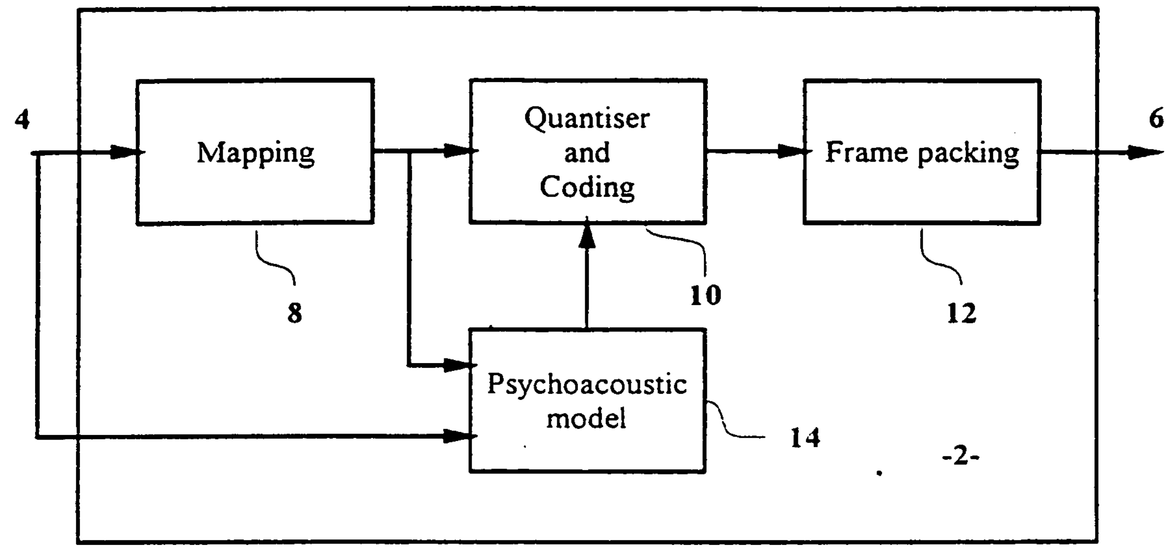

Figure 1

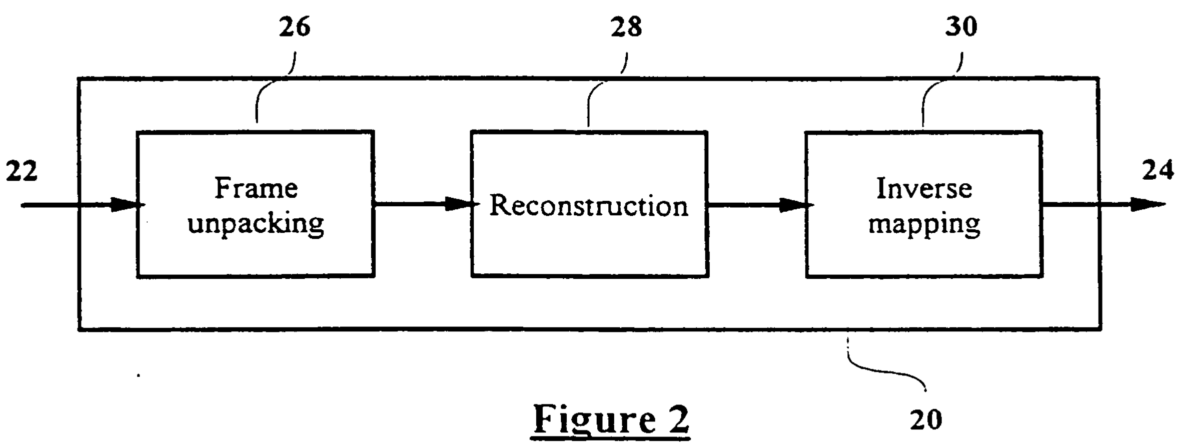

Figure 2

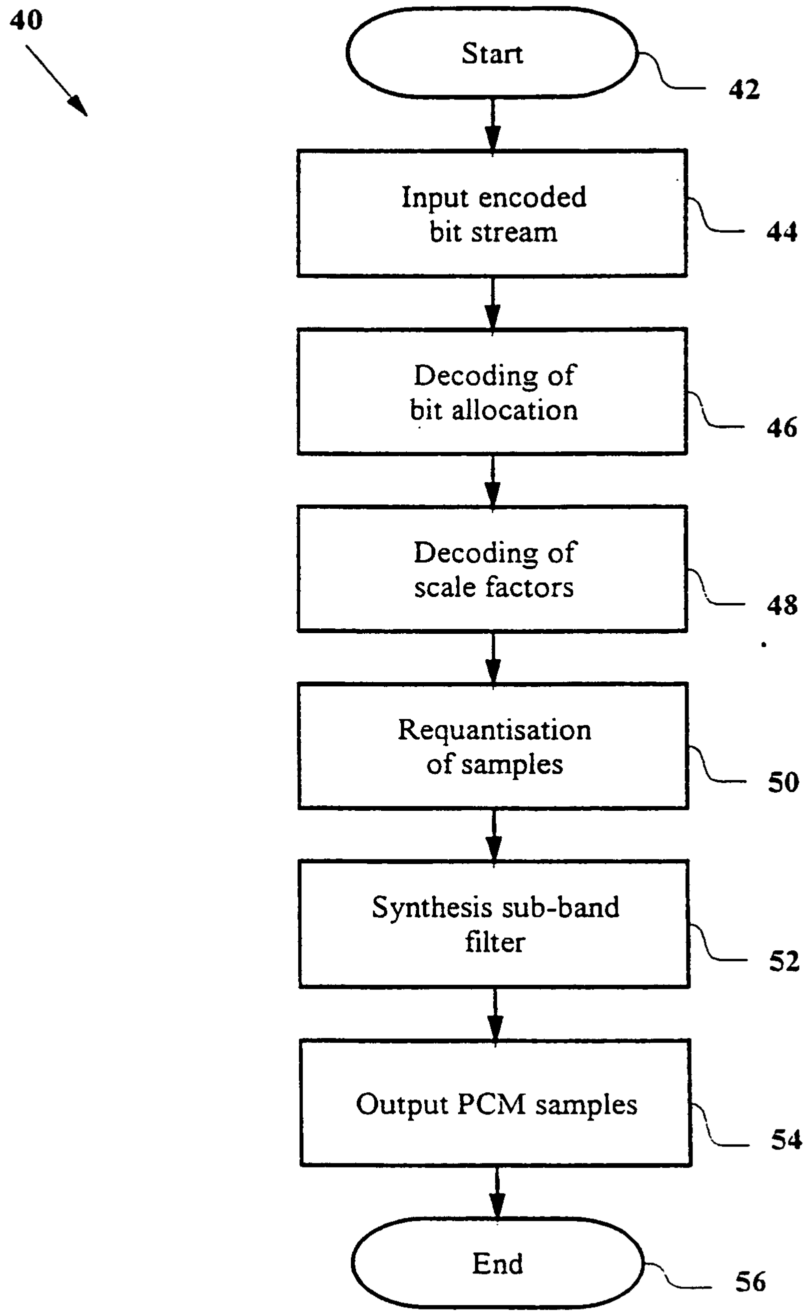
40
Start
42
Input encoded bit stream
44
Decoding of bit allocation
46
Decoding of scale factors
48
Requantisation of samples
50
Synthesis sub-band filter
52
Output PCM samples
54
End
56

Figure 3

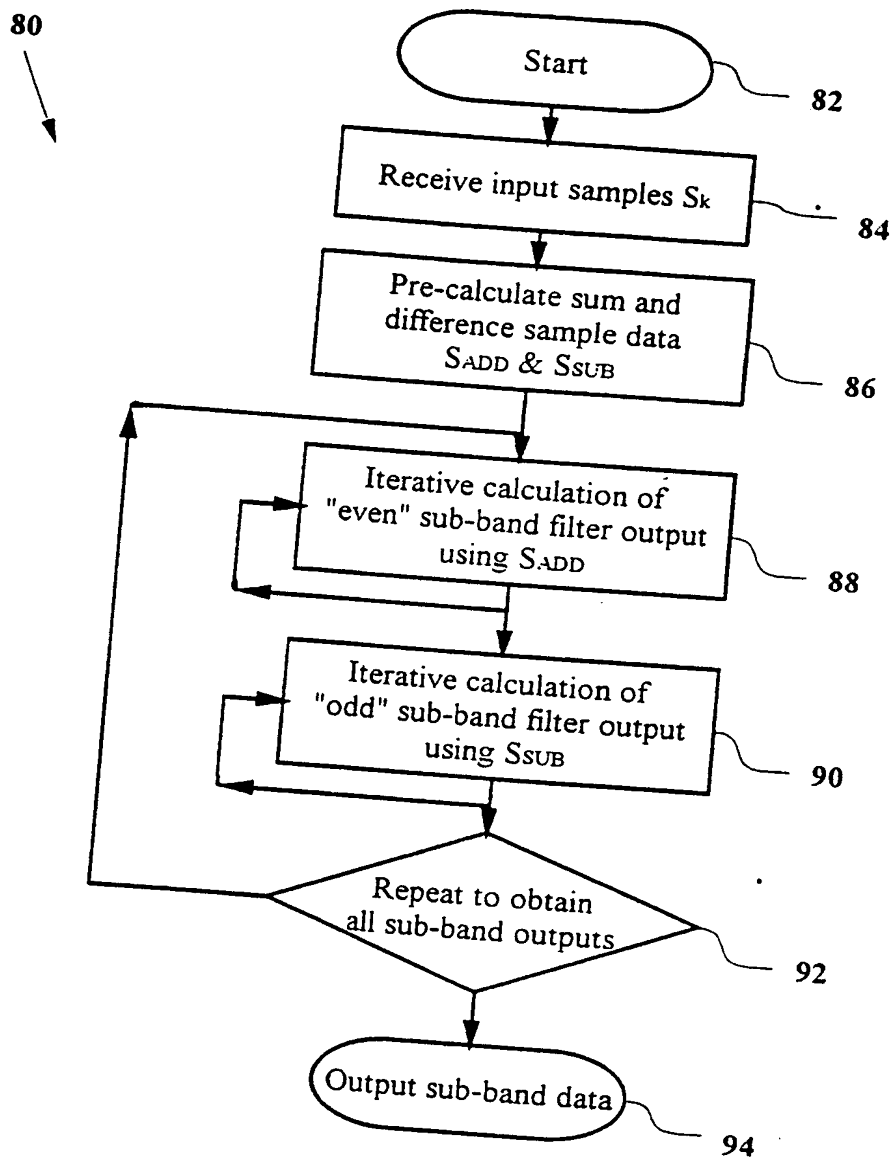
80
Start
82
Receive input samples Sk
84
Pre-calculate sum and difference sample data SADD & SSUB
86
Iterative calculation of "even" sub-band filter output using SADD
88
Iterative calculation of "odd" sub-band filter output using SSUB
90
Repeat to obtain all sub-band outputs
92
Output sub-band data
94

Figure 4

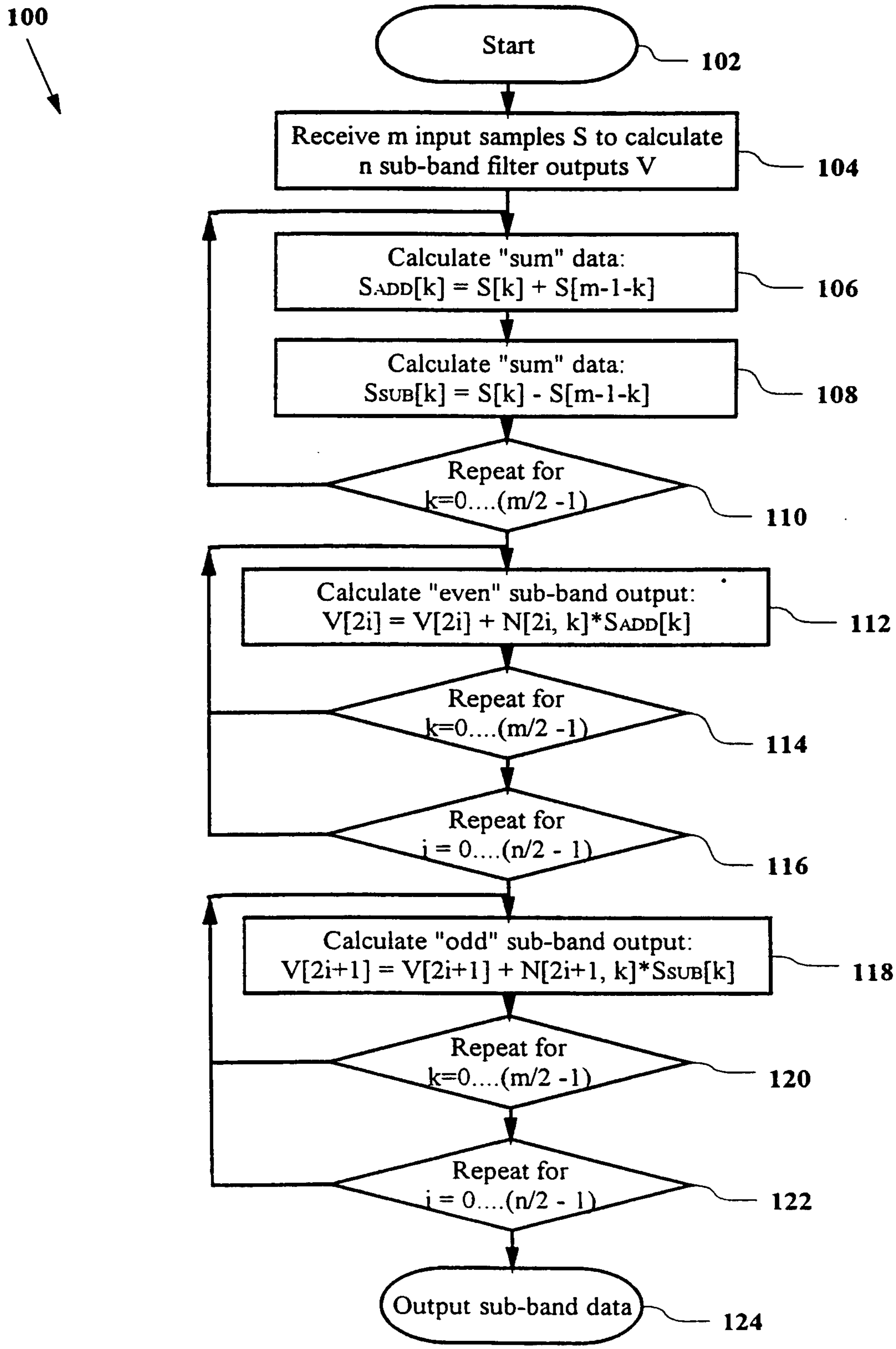
100
Start
102
Receive m input samples S to calculate n sub-band filter outputs V
104
Calculate "sum" data: SADD[k] = S[k] + S[m-1-k]
106
Calculate "sum" data: SSUB[k] = S[k] - S[m-1-k]
108
Repeat for k=0....(m/2 -1)
110
Calculate "even" sub-band output: V[2i] = V[2i] + N[2i, k]*SADD[k]
112
Repeat for k=0....(m/2 -1)
114
Repeat for i = 0....(n/2 - 1)
116
Calculate "odd" sub-band output: V[2i+1] = V[2i+1] + N[2i+1, k]*SSUB[k]
118
Repeat for k=0....(m/2 -1)
120
Repeat for i = 0....(n/2 - 1)
122
Output sub-band data
124

Figure 5

INTERNATIONAL SEARCH REPORT

International Application No
PCT/SG 97/00037

A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04H1/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04H

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	EP 0 506 111 A (MITSUBISHI ELECTRIC CORP) 30 September 1992	1
A	see page 2, line 1 - page 5, line 16; claim 1	8,11
A	US 5 181 183 A (MIYAZAKI TAKASHI) 19 January 1993 see column 1, line 1 - column 2, line 27; claim 1; figures 1-3	1,8,11
A	EP 0 590 790 A (SONY CORP) 6 April 1994 see page 2, line 1 - line 31; claim 1	1,8,11
A	US 5 257 213 A (LEE SANG-YOOK ET AL) 26 October 1993 see column 2, line 55 - column 3, line 52; claim 1	1,8,11

-/--

[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

° Special categories of cited documents :

"A" document defining the general state of the art which is not considered to be of particular relevance

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"O" document referring to an oral disclosure, use, exhibition or other means

"P" document published prior to the international filing date but later than the priority date claimed

"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention

"X" document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone

"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art.

"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
3 June 1998	12/06/1998

Name and mailing address of the ISA	Authorized officer
European Patent Office, P.B. 5818 Patentlaan 2 NL - 2280 HV Rijswijk Tel. (+31-70) 340-2040, Tx. 31 651 epo nl, Fax: (+31-70) 340-3016	De Haan, A.J.

1

INTERNATIONAL SEARCH REPORT

International Application No
PCT/SG 97/00037

C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT		
Category	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	EP 0 564 089 A (AMERICAN TELEPHONE & TELEGRAPH) 6 October 1993 see page 2, line 1 - page 3, line 57; claim 1; figure 1 -----	1,8,11

INTERNATIONAL SEARCH REPORT

Information on patent family members

International Application No
PCT/SG 97/00037

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		JP 6029859 A	04-02-1994
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